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IN REPLY REFER TO: Joint Interoperability Test Command (JTE) **17 Feb 15**

MEMORANDUM FOR DISTRIBUTION Revision 1

SUBJECT: Extension of the Joint Interoperability Certification of the Cisco Enterprise Session Controller (ESC) 8

References: (a) Department of Defense Instruction 8100.04, "DoD Unified Capabilities (UC)," 9 December 2010
(b) Office of the Department of Defense Chief Information Officer, “Department of Defense Unified Capabilities Requirements 2013, Errata 1,” 1 July 2013
(c) through (e), see Enclosure 1

1. **Certification Authority.** Reference (a) establishes the Joint Interoperability Test Command (JITC) as the Joint Interoperability Certification Authority for the UC products.

2. **Conditions of Certification.** The Cisco ESC 8; hereinafter referred to as the System Under Test (SUT), meets the critical requirements of the Unified Capabilities Requirements (UCR), Reference (b), and is certified for joint use as an ESC in Type 1, 2, and 3 environments and as a Local Session Controller (LSC) with the conditions described in Table 1. This certification expires upon changes that affect interoperability, but no later than 27 June 2017, which is three years from the date of the original UC Approved Products List (APL) memorandum. Desktop Review (DTR) 4 was requested to update the SUT Cisco Webex Meeting Server from Release 2.0 to 2.5 and remove the Cisco MeetingPlace Server from the SUT. See paragraph 4 for the test details.

Table 1. Conditions

Condition	Operational Impact	Remarks
UCR Waivers		
None.		
Conditions of Fielding		
None.		
Open Test Discrepancies		
The SUT video end instruments include H.323 proprietary ROUTINE only end instruments depicted in Table 4. Additionally the SUT includes a Jabber client that offers video and voice; however, during the original test the Jabber client was certified for audio only as a soft phone and for XMPP IM&P as an XMPP client.	None	CLOSED See note 1.
Per the vendor's LoC, the SUT does not display weighted Terminal Coupling Loss (TCLw) and equipment impairment factor in their call detail record (CDR).	Minor	See note 2.
Per the vendor's LoC, the does not fully meet separate video and voice ASAC counts.	Minor	See note 2.

Table 1. Conditions (continued)

Condition	Operational Impact	Remarks
Open Test Discrepancies (continued)		
The SUT does not properly handle signaling events when setting up an inter-switch V.150 secure call with Avaya Communication Manager (CM) 6.0.	Minor	See note 3.
Per the vendor's LoC, the SUT proprietary video EI does not provide the ability to enable or disable the transmission destination unreachable msg.	Minor	See note 2.
Per the vendor's LoC, the SUT fails to immediately divert all precedence above routine calls placed to ROEIs. The SUT diverts only when the ROEI is busy if it is idle it will offer the call and divert if not answered.	Minor	See note 2.
During the original test, video calls between SUT H.323 PEIs (C90/EX90/SX20) and other UC video endpoints dropped at approximately 30 minutes.	None	CLOSED See note 4.
The SUT fails to answer with correct payload number per RFC 3264. Instead, of responding to the V.150.1 payload numbers in an SDP, offer the SUT always responds with payload number of 118 and 120 for State Signaling Events (SSE) and Simple Packet Relay Transport (SPRT) respectively which prevents successful secure call attempts.	Minor	See note 3.
Per the vendor's LoC, the SUT does not support an AS-SIP ESC to EI signaling interface.	Minor	See note 3.
Per the vendor's LoC, the SUT supports Primary Rate Interface requirement to be in compliance with ANSI T1.619-1992 and T1.619a-1994 with following exception, NFAS is not supported.	Minor	See note 2.
Per the vendor's LoC, the SUT does not support Public Key Infrastructure Requirement IA-049030.	Minor	See note 5.
Per the vendor's LoC, the SUT does not support Confidentiality requirement IA-069040.	Minor	See note 5.
The SUT 9951/9971 voice/video SIP ROEIs do not fully support inter-enclave hold feature while video enabled.	Minor	See note 6.
Per the vendor's LoC, the SUT does not support a persistent TLS connection between AEIs and the enclave fronted SBC because the SUT does not support AEIs.	Minor	See note 3.
Per the vendor's LoC, the SUT video conferencing system does not support all required audio codecs. The SUT does not support the G.723.1 audio codec.	Minor	See note 2.
Per the vendor's LoC, the SUT partially complies with the EDS gateway requirements per SCM-005300.	Minor	CLOSED See note 7.
When the SUT MCU 5320 places an outbound video call to other SUT C90 and SX20 video endpoints in either environment 1 or environment 2, the call drops at exactly 15 minutes.	Minor	See note 3.
The SUT SX20 and C90 configured on environment 2 are not able to establish two-way video calls with the Polycom RMX UCCS. The SUT SX20 configured on environment 1 or environment 2 is not able to establish two-way video calls with the Vidyo UCCS. These anomalies occur when the SX20 and C90 are registered to the ESC Environments and do not occur when these endpoints are registered to the LSC.	Minor	See note 3.
Per the vendor's LoC, the SUT does not correctly respond to stream errors. Instead of responding with a stream error and closing the stream, the server terminates the connection non-gracefully.	Minor	See note 3.
Per the vendor's LoC, the SUT does not generate a new Client-to-Server Stream. Server reuses the old stream ID instead of generating a new stream ID.	Minor	See note 3.
Per the vendor's LoC, the SUT does not include empty element in its advertisement of the SASL.	Minor	See note 8.
Per the vendor's LoC, the SUT does not fully comply with SASL failure requirements. The SUT does not comply with requirements IM-000710, IM-000720, and IM-000730.	Minor	See note 3.
Per the vendor's LoC, the SUT does not fully meet deleting a roster item requirement. The SUT does not comply with requirements IM-001310 and IM-001320.	Minor	See note 3.
Per the vendor's LoC, the SUT partially complies with rules for Server Processing of Outbound Subscription Requests. The SUT does not comply with requirement IM-001350. Server sends presence type "unsubscribed" with status Not Found.	Minor	See note 3.
Per the vendor's LoC, the SUT partially complies with the rules for server processing of outbound subscription cancellation. The SUT partially complies with requirement IM-001500. Upon receiving the outbound subscription cancellation, the contact's server does not send a presence stanza of type "unavailable" from all of the contacts online resources to the user.	Minor	See note 2.
Per the vendor's LoC, the SUT partially complies with the rules for server processing of inbound unsubscribe. The SUT partially complies with requirement IM-001540.	Minor	See note 2.
Per the vendor's LoC, the SUT does not comply with server generation of inbound presence probe.	Minor	See note 3.
The SUT Unified Presence Server establishes SASL external authentication with the incorrect domain name.	Minor	See note 3.

Table 1. Conditions (continued)

Condition	Operational Impact	Remarks
Open Test Discrepancies (continued)		
The SUT does not comply to the requirements in XMPP Extension XEP-0045 (multi-user chat). The SUT does not host or participate in multi-user chat/chat rooms as required by the reference.	Minor	See note 3.
The SUT Jabber Video Client when calling the Polycom Group series video EI has 1-way audio.	Minor	CLOSED See note 1.
The SUT does not support Local RTS Database (LRDB).	Minor	See note 2.
The SUT does not support Master RTS Database (MRDB).	Minor	See note 2.

NOTES:
1. During the original test, the SUT Jabber Video Client when calling the Polycom Group series video EI had 1-way audio. In the original certification, the Jabber client was certified for audio only as a soft phone and for XMPP IM&P as an XMPP client. The video discrepancy was fixed and the video capability of Jabber client was successfully tested during the DTR 2 test window. See the deployment guide for the configuration change.
2. DISA has adjudicated this discrepancy as minor and stated the intent to change this requirement in the next version of the UCR.
3. DISA has accepted the vendor's POA&M and has adjudicated this discrepancy as minor.
4. During the original test, video calls between SUT H.323 PEIs (C90/EX90/SX20) and other UC video endpoints dropped at approximately 30 minutes. This discrepancy with call drops at 30 minutes was fixed and successfully tested with DTR 1, which included updated VCS software release x8.1.1.
5. DISA has adjudicated this discrepancy as minor and stated the intent to remove this requirement from the UCR and apply it to a DoD STIG.
6. DISA has accepted the vendor's POA&M and has adjudicated this discrepancy as minor. In addition, the 9951/9971 voice/video SIP ROEI is not covered under this certification.
7. This discrepancy applies only to the SUT configured as an ESC. DISA has accepted the vendor's POA&M and has adjudicated this discrepancy as minor. In the interim, the LiteScape EDS Gateway has been successful with posting their LiteScape EDS gateway on the UC APL under tracking number 1412803. LiteScape is certified on the UC APL only with the SUT. The SUT now meets the EDS ESC minimum essential interoperability requirements with the LiteScape EDS Gateway.
8. DISA has adjudicated this discrepancy as minor.

LEGEND:

AEI	AS-SIP End Instrument	PEI	Proprietary End Instrument
ANSI	American National Standards Institute	POA&M	Plan of Action and Milestones
APL	Approved Products List	RFC	Request for Comments
ASAC	Assured Services Admission Control	ROEI	ROUTINE Only End Instrument
AS-SIP	Assured Services Session Initiation Protocol	SASL	Simple Authentication and Security Layer
CUPS	Cisco Unified Presence Server	SBC	Session Border Controller
DISA	Defense Information System Agency	SDP	Session Description Protocol
DN	Directory Number	SIP	Session Initiation Protocol
DTR	Desktop Review	SUT	System Under Test
EDS	Enterprise Directory Services	STIG	Security Technical Implementation Guide
EI	End Instrument	TLS	Transport Layer Security
ESC	Enterprise Session Controller	UC	Unified Capabilities
ID	identification	UCCS	Unified Capabilities Conference System
IM/P	Instant Messaging/Presence	UCR	Unified Capabilities
LoC	Letter of Compliance	VCS	Video Communication Server
MCU	Multipoint Control Unit	XMPP	Extensible Messaging and Presence Protocol
NFAS	Non Facility Associated Signaling		

3. **Interoperability Status.** Table 2 provides the SUT interface interoperability status and Table 3 provides the Capability Requirements (CR) and Functional Requirements (FR) status. Table 4 provides the UC APL product summary.

Table 2. Interface Status

Interface	Threshold CR/FR Requirements (See note.)	Status	Remarks
Network Management Interfaces			
10BaseT (R)	4, 6, 9, 13, 16, 20, 21, 23, 24	Certified	The SUT met the critical CRs and FRs for the IEEE 802.3i interface.
100BaseT (R)	4, 6, 9, 13, 16, 20, 21, 23, 24	Certified	The SUT met the critical CRs and FRs for the IEEE 802.3u interface.
1000BaseT (C)	4, 6, 9, 13, 16, 20, 21, 23, 24	Certified	The SUT met the critical CRs and FRs for the IEEE 802.3ab interface.
Network Interfaces (Line and Trunk)			
10BaseT (R)	1, 5, 6, 7, 8, 10, 11, 13, 14, 15, 17, 18, 19, 20, 21, 22, 23, 24, 25	Certified	The SUT met the critical CRs and FRs for the IEEE 802.3i interface with the SUT PEIs and softphones.
100BaseT (R)	1, 5, 6, 7, 8, 10, 11, 13, 14, 15, 17, 18, 19, 20, 21, 22, 23, 24, 25	Certified	The SUT met the critical CRs and FRs for the IEEE 802.3u interface with the SUT PEIs and softphones.
1000BaseT (R)	1, 5, 6, 7, 8, 10, 11, 13, 14, 15, 17, 18, 19, 20, 21, 22, 23, 24, 25	Certified	The SUT met the critical CRs and FRs for the IEEE 802.3ab interface with the SUT PEIs and softphones.
2-wire analog (R)	1, 8, 15, 17, 19, 20, 21, 22, 23	Certified	The SUT met the critical CRs and FRs for the 2-wire analog interface with the SUT 2-wire secure and non-secure analog instruments.
ISDN BRI (C)	1, 8, 15, 17, 19, 20, 21, 22, 23	Not Tested	The SUT offers this interface; however, it was not tested because it does not support Assured Services and is not required for an ESC.
Legacy Interfaces (External)			
10BaseT (C)	2, 3, 5, 6, 7, 8, 11, 13, 18, 20, 21, 23, 24, 25	Certified	The SUT met the critical CRs/FRs for IEEE 802.3i for the AS-SIP trunk.
100BaseT (C)	2, 3, 5, 6, 7, 8, 11, 13, 18, 20, 21, 23, 24, 25	Certified	The SUT met the critical CRs/FRs for IEEE 802.3u for the AS-SIP trunk.
1000BaseT (C)	2, 3, 5, 6, 7, 8, 11, 13, 18, 20, 21, 23, 24, 25	Certified	The SUT met the critical CRs/FRs for IEEE 802.3ab for the AS-SIP trunk.
ISDN T1 PRI (ANSI T1.619a) (R)	3, 9, 12, 14, 22, 20, 21, 23	Certified	The SUT met the critical CRs/FRs. This interface provides legacy DSN and TELEPORT connectivity.
ISDN T1 PRI NI-2 (R)	3, 9, 12, 14, 22, 20, 21, 23	Certified	The SUT met the critical CRs/FRs. This interface provides PSTN connectivity.
T1 CCS7 (ANSI T1.619a) (C)	3, 9, 12, 14, 20, 21, 22, 23	Not Tested	The SUT does not support this conditional interface.
T1 CAS (C)	3, 9, 12, 14, 20, 21, 22, 23	Certified	The SUT met threshold CRs/FRs for DTMF.
E1 PRI (ITU-T Q.955.3) (C)	3, 9, 12, 14, 20, 21, 22, 23	Certified	The SUT met the critical CRs/FRs. This interface provides OCONUS MLPP connectivity in ETSI-compliant countries.
E1 PRI (ITU-T Q.931) (C)	3, 9, 12, 14, 20, 21, 22, 23	Certified	The SUT met the critical CRs/FRs. This interface provides OCONUS connectivity in ETSI-compliant countries.
NOTE: The SUT high-level CR and FR ID numbers depicted in the Threshold CRs/FRs column can be cross-referenced in Table 3. These high-level CR/FR requirements refer to a detailed list of requirements provided in Reference (c), Enclosure 3.			

Table 2. Interface Status (continued)

LEGEND:

10BaseT	10 Mbps Ethernet	IEEE	Institute of Electrical and Electronics Engineers
100BaseT	100 Mbps Ethernet	ISDN	Integrated Services Digital Network
1000BaseT	1000 Mbps Ethernet	ITU-T	International Telecommunication Union - Telecommunication Standardization Sector
ANSI	American National Standards Institute		
AS-SIP	Assured Services Session Initiation Protocol	Mbps	Megabits per second
BRI	Basic Rate Interface	MLPP	Multi-Level Precedence and Preemption
C	Conditional	NI-2	National ISDN Standard 2
CAS	Channel Associated Signaling	OCONUS	Outside the Continental United States
CCS7	Common Channel Signaling Number 7	PEI	Proprietary End Instrument
CR	Capability Requirement	PRI	Primary Rate Interface
DSN	Defense Switched Network	PSTN	Public Switched Telephone Network
DTMF	Dual Tone Multi-Frequency	Q.931	Signaling Standard for ISDN
E1	European Basic Multiplex Rate (2.048 Mbps)	Q.955.3	ISDN Signaling Standard for E1 MLPP
ESC	Enterprise Session Controller	R	Required
ETSI	European Telecommunications Standards Institute	SS7	Signaling System 7
FR	Functional Requirement	T1	Digital Transmission Link Level 1 (1.544 Mbps)
ID	Identification	T1.619a	SS7 and ISDN MLPP Signaling Standard for T1

Table 3. SUT Capability Requirements and Functional Requirements Status

CR/FR ID	UCR Requirement (High-Level) (See note 1.)	UCR 2013 Reference	Status
1	Voice Features and Capabilities (R)	2.2	Partially Met (See note 2.)
2	Assured Services Admission Control (R)	2.3	Met
3	Signaling Protocols (R)	2.4	Met
4	Registration and Authentication (R)	2.5	Met
5	SC and SS Failover and Recovery (R)	2.6	Met
6	Product Interface (R)	2.7	Met
7	Product Physical, Quality, and Environmental Factors (R)	2.8	Met
8	End Instruments (including tones and announcements) (R)	2.9	Partially Met (See note 2.)
9	Session Controller (R)	2.10	Met
10	AS-SIP Gateways (C)	2.11	Met (See note 3.)
11	Enterprise UC Services (R)	2.12	Partially Met (See notes 2 4, and 5.)
12	Call Connection Agent (R)	2.14	Met
13	CCA Interaction with Network Appliances and Functions (R)	2.15	Met
14	Media Gateway (R)	2.16	Met
15	Worldwide Numbering & Dialing Plan (R)	2.18	Met
16	Management of Network Devices (R)	2.19	Partially Met (See note 2.)
17	V.150.1 Modem Relay Secure Phone Support (R)	2.20	Partially Met (See note 2.)
18	Requirements for Supporting AS-SIP Based Ethernet Devices for Voicemail Systems (C)	2.21	Not Tested
19	Local Attendant Console Features (O)	2.22	Not Tested
20	MSC and SSC (O)	2.23	Not Tested (See note 6.)
21	MSC, SSC, and Dynamic ASAC Requirements in Support of Bandwidth-constrained links (O)	2.24	Not Tested (See note 7.)
22	Other UC Voice (R)	2.25	Partially Met (See note 2.)
23	Information Assurance Requirements (R)	4	Partially Met (See notes 2 and 7.)
24	IPv6 Requirements (R)	5	Partially Met (See note 2.)
25	Assured-Services (AS) Session Initiation Protocol (SIP) (AS-SIP 2013) (R)	AS-SIP	Partially Met (See note 2.)

Table 3. SUT Capability Requirements and Functional Requirements Status (continued)

NOTES:
1. The annotation of 'required' refers to a high-level requirement category. The applicability of each sub-requirement is provided in Reference (c), Enclosure 3.
2. The SUT met the requirements with the exceptions noted in Table 1. DISA adjudicated these exceptions as minor.
3. During the original test, video calls between SUT H.323 PEIs (C90/EX90/SX20) and other UC video endpoints dropped at approximately 30 minutes. This discrepancy was fixed and successfully tested with DTR 1, which included VCS software release x8.1.1.
4. These requirements apply specifically to an Enterprise Session Controller.
5. During the original test, the SUT Jabber Video Client when calling the Polycom Group series video EI had 1-way audio. In the original certification, the Jabber client was certified for audio only as a soft phone and for XMPP IM&P as an XMPP client. The video discrepancy was fixed and the video capability of Jabber client was successfully tested during the DTR 2 test window. See the deployment guide for the configuration change.
6. This optional requirement applies specifically to a Local Session Controller.
7. Security is tested by DISA-led Information Assurance test teams and the results published in a separate report, Reference (e).

LEGEND:

AS-SIP	Assured Services Session Initiation Protocol	O	Optional
C	Conditional	PEI	Proprietary End Instrument
CCA	Call Connection Agent	R	Required
CR	Capability Requirement	SC	Session Controller
DISA	Defense Information System Agency	SS	Softswitch
DTR	Desktop Review	SUT	System Under Test
FR	Functional Requirement	UC	Unified Capabilities
ID	Identification	UCR	Unified Capabilities Requirements
IPv6	Internet Protocol version 6	VCS	Video Communication Server

Table 4. UC APL Product Summary

Product Identification			
Product Name	Cisco Enterprise Session Controller (ESC) 8		
Software Release	8		
UC Product Type(s)	Enterprise Session Controller (ESC) or Local Session Controller		
Product Description	Enterprise Session Controller for Type 1, 2, and 3 Environments or as a Local Session Controller		
Product Components (See note 1.)	**Component Name (See notes 2 and 3.)**	**Version**	**Remarks**
Unified Communications Manager	Cisco Unified Communications Manager	8.6	
Session Management Edition	Cisco Session Management Edition	8.6	
Unified Communications Manager	**Cisco Unified Communications Manager**	8.6	
Cisco Unity Connection	Cisco Unity Connection	8.6	
Cisco Unified Presence Server	Cisco Unified Presence Server	8.6	
Cisco Webex Meeting Server	Cisco Webex Meeting Server	2.5 (See note 4.)	
E911 management system	RedSky E911 Management System	6.3.1	See note 5.
Interworking Gateway	IWG on 3925 ISR G2, IWG on 3925E ISR G2, **IWG on 3945 ISR G2,** IWG on 3945E ISR G2	IOS 15.2(4)M5	
Session Border Controller	SBC on 3925 ISR G2, SBC on 3925E ISR G2, **SBC on 3945 ISR G2,** SBC on 3945E ISR G2	IOS 15.2(4)M5	
Session Border Controller	**SBC on ISR 4451-X Router**	IOS-XE 3.11	
Session Border Controller	**SBC on ASR 1002,** SBC on ASR 1002-X, SBC on ASR 1004, SBC on ASR 1006	IOS-XE 3.11	
Voice Gateway	2901 ISR G2, 2911 ISR G2, 2921 ISR G2, 2951 ISR G2, 3925 ISR G2, 3925E ISR G2, **3945 ISR G2,** 3945E ISR G2	IOS 15.2(4)M5	
Analog Voice Gateway	VG350 Analog Voice Gateway	IOS 15.2(4)M5	
Jabber	Cisco Jabber for Windows	9.2	See note 6.
IP Phone	Unified IP Phone 6901	9.2.1	
IP Phone	Unified IP Phone 6911	9.2.1	

Table 4. UC APL Product Summary (continued)

Product Components (See note 1.)	Component Name (See notes 2 and 3.)	Version	Remarks
IP Phone	Unified IP Phone 6911	9.2.1	
IP Phone	Unified IP Phone 6921	9.2.1	
IP Phone	Unified IP Phone 6941	9.2.1	
IP Phone	Unified IP Phone 6945	9.2.1	
IP Phone	Unified IP Phone 6961	9.2.1	
IP Phone	Unified IP Phone 7821	10.1.1.9	
IP Phone	Unified IP Phone 7841	10.1.1.9	
IP Phone	Unified IP Phone 7861	10.1.1.9	
IP Phone	Unified IP Phone 7906G	9.3.1	
IP Phone	Unified IP Phone 7911G	9.3.1	
IP Phone	Unified IP Phone 7931G	9.3.1	
IP Phone	Unified IP Phone 7941G	9.3.1	
IP Phone	Unified IP Phone 7941G-GE	9.3.1	
IP Phone	Unified IP Phone 7942G	9.3.1	
IP Phone	Unified IP Phone 7945G	9.3.1	
IP Phone	Unified IP Phone 7961G	9.3.1	
IP Phone	Unified IP Phone 7961G-GE	9.3.1	
IP Phone	Unified IP Phone 7962G	9.3.1	
IP Phone	Unified IP Phone 7965G	9.3.1	
IP Phone	Unified IP Phone 7970G	9.3.1	
IP Phone	Unified IP Phone 7971G	9.3.1	
IP Phone	Unified IP Phone 7975G	9.3.1	
IP Phone	Unified IP Phone Expansion Module 7915	Not Applicable	
IP Phone	Unified IP Phone Expansion Module 7916	Not Applicable	
IP Conference Phone	IP Conference Station 8831	9.3.3.5	
IP Phone	Unified IP Phone 8961	9.4.1	
IP Phone	Unified IP Phone 9951	9.4.1	See note 7.
IP Phone	Unified IP Phone 9971	9.4.1	See note 7.
IP Phone Expansion Module	Unified IP Color Key Expansion Module	Not Applicable	
Secure Phone	CIS Secure DTD-7965-TSGB	9.3.1	
Secure Phone	CIS Secure DTD-7962-TSG-01	9.3.1	
Secure Phone	CIS Secure DTD-7962-T2	9.3.1	
Secure Phone	Telecore 2151	2AE-00199-0301	
Video Teleconference	TelePresence Video Communication Server (VCS)	X8.1.1 (See note 8.)	
Video Teleconference	TelePresence QuickSet C20	TC7.1.1	
Video Teleconference	TelePresence Codec C40, TelePresence Codec C60, **TelePresence Codec C90**	TC7.1.1	
Video Teleconference	TelePresence EX60, TelePresence EX90	TC7.1.1	
Video Teleconference	TelePresence MX200, TelePresence MX300	TC7.1.1	
Video Teleconference	**TelePresence SX20 QuickSet,** TelePresence MX300 G2	TC7.1.1	
Video Teleconference	**TelePresence 5300 MCU**	4.4(1.68)	
Common Access Card/Single sign-on solution	**OpenAM**	11.0 (See note 9.)	
Common Access Card support for WebEx Meeting Server	Cisco ASA	8.4(3)	

Table 4. UC APL Product Summary (continued)

NOTES:
1. The detailed component and subcomponent list is provided in Reference (c), Enclosure 3.
2. Components bolded and underlined were tested by JITC. The other components in the family series were not tested but are also certified for joint use. JITC certifies those additional components because they utilize the same software and similar hardware and JITC analysis determined them to be functionally identical for interoperability certification purposes.
3. A comprehensive list of supported hardware configurations can be found by selecting the "Cisco Unified Communications on the Cisco Unified Computing System" link at the following URL: www.cisco.com/go/swonly.
4. The SUT Cisco Webex Meeting Server (CWMS) was updated from Release 2.0 to 2.5 with DTR 4. This CWMS update includes support for preset conferencing and security revisions that included Single Sign On capability via Security Assertion Markup Language verison 2.0, which were successfully tested with DTR 4. In addition, DTR 4 documents the removal of the Cisco MeetingPlace Server from the SUT.
5. The SUT is certified with any RedSky E911 Management system or other E911 Management system listed on the UC APL and certified with the Cisco UCM. E911 management is not required for an LSC.
6. During the original test, the SUT Jabber Video Client when calling the Polycom Group series video EI had 1-way audio. In the original certification, the Jabber client was certified for audio only as a soft phone and for XMPP IM&P as an XMPP client. The video discrepancy was fixed and the video capability of Jabber client was successfully tested during the DTR 2 test window. See the deployment guide for the configuration change.
7. The SUT 9951/9971 voice/video SIP ROEIs do not fully support inter-enclave hold feature while video enabled. DISA has accepted the vendor’s POA&M and has adjudicated this discrepancy as minor. In addition, the 9951/9971 voice/video SIP ROEI is not covered under this certification.
8. The VCS release was updated from x7.2.2 to x8.1.1 with DTR 1.
9. The OpenAM version was updated from 9.5.5 to 11.0 with DTR 5.

LEGEND:

APL	Approved Products List	MCU	Multipoint Conference Unit
DISA	Defense Information System Agency	POA&M	Plan of Action and Milestones
DTR	Desktop Review	ROEI	ROUTINE Only End Instrument
G2	Generation 2	SBC	Session Border Controller
IM/P	Instant Messaging/Presence	SIP	Session Initiation Protocol
IP	Internet Protocol	UC	Unified Capabilities
ISR	Integrated Services Router	UCM	Unified Communications Manager
IWG	Interworking Gateway	VCS	Video Communication Server
JITC	Joint Interoperability Test Command	XMPP	Extensible Messaging and Presence Protocol

4. **Test Details.** The extension of this certification is based upon DTR 4. The original certification, documented in Reference (c), is based on interoperability testing, DISA adjudication of open test discrepancy reports (TDRs), review of the vendor’s Letters of Compliance (LoC), and DISA Certifying Authority (CA) Recommendation for inclusion on the UC APL. Testing was conducted under UCCO Tracking Number 1108301 from 11 July through 5 August 2011 on the SUT as an LSC. Additional testing of the LSC under UCCO Tracking Number 1108301 was conducted for Desktop Reviews and documented in extensions to the original certification. Testing was conducted from 7 April through 9 May 2014 on the Cisco UCM as an ESC. The data from the LSC test is included in this certification. The test procedures derived from the UCR Reference (b) using test procedures derived from Reference (d) were used to validate the deltas between a Local Session Controller (LSC) and an ESC. Review of the vendor’s LoC was completed on 7 April 2014. DISA adjudication of outstanding TDRs was completed on 10 June 2014. Information Assurance (IA) testing was conducted by DISA-led Information Assurance test teams and the results are published in a separate report, Reference (e). This DTR was requested to update the SUT Cisco Webex Meeting Server (CWMS) from Release 2.0 to 2.5 and to remove the Cisco MeetingPlace Server from the SUT. The CWMS update includes security modifications that include Single Sign On (SSO) capability via Security Assertion Markup Language (SAML) version 2.0 as well as support for preset conferencing. JITC determined that IA and interoperability Verification and Validation (V&V) testing was required. The IA V&V testing was successfully completed from 5 through 16 January 2015 and the results published in a separate report, Reference (e). JITC conducted interoperability V&V testing with no findings from 19 January through 6 February 2015.

Therefore, JITC approves this DTR. Enclosure 2 provides a list of errata changes to this certification since the original signature date.

5. **Additional Information.** JITC distributes interoperability information via the JITC Electronic Report Distribution (ERD) system, which uses Sensitive but Unclassified IP Data (formerly known as NIPRNet) e-mail. Interoperability status information is available via the JITC System Tracking Program (STP). STP is accessible by .mil/.gov users at https://stp.fhu.disa.mil/. Test reports, lessons learned, and related testing documents and references are on the JITC Joint Interoperability Tool (JIT) at https://jit.fhu.disa.mil/. Due to the sensitivity of the information, the Information Assurance Accreditation Package (IAAP) that contains the approved configuration and deployment guide must be requested directly from the Unified Capabilities Certification Office (UCCO), e-mail: disa.meade.ns.list.unified-capabilities-certification-office@mail.mil. All associated information is available on the DISA UCCO website located at http://www.disa.mil/Services/Network-Services/UCCO.

6. **Point of Contact (POC).** The JITC point of contact is Mr. Joseph Schulte, commercial telephone (520) 538-5100, DSN telephone 879-5100, FAX DSN 879-4347; e-mail address joseph.t.schulte.civ@mail.mil; mailing address Joint Interoperability Test Command, ATTN: JTE (Mr. Joseph Schulte) P.O. Box 12798, Fort Huachuca, AZ 85670-2798. The UCCO tracking number for the SUT is 1331201.

FOR THE COMMANDER:

Bradley A. Clark

2 Enclosures a/s

for RIC HARRISON
Chief
Networks/Communications and UC Portfolio

Distribution (electronic mail):
DoD CIO
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USD(AT&L)
ISG Secretariat, DISA, JTA
U.S. Strategic Command, J665
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DISA/TEMC
DIA, Office of the Acquisition Executive
NSG Interoperability Assessment Team
DOT&E, Netcentric Systems and Naval Warfare
Medical Health Systems, JMIS IV&V
HQUSAISEC, AMSEL-IE-IS
UCCO

ADDITIONAL REFERENCES

(c) Joint Interoperability Test Command, Memo, JTE, "Joint Interoperability Certification of the Cisco Enterprise Session Controller (ESC) 8," 13 June 2014
(d) Joint Interoperability Test Command, "Enterprise Session Controller (ESC) Test Procedures for Unified Capabilities Requirements (UCR) 2013," Draft
(e) Joint Interoperability Test Command, "Information Assurance (IA) Findings Summary For Cisco ESC 8 (Tracking Number 1331201)," Draft

Joint Interoperability Certification Revision History (continued)

Revision	Date	Approved By	Comments
NA	17 February 2015	Bradley Clark	This is the original Extension of the Joint Interoperability Certification.
1	5 May 2015	Joseph Schulte	The following changes were made to update the 79xx, 8961, and 99x1 IP phones and the 8831 IP Conference Station from version 9.2.1 to the version tested. • Memo, Page 7, Table 4. The 79xx series IP phones were updated from version 9.2.1 to 9.3.1. • Memo, Page 7, Table 4. The 8831 IP Conference Station was updated from version 9.2.1 to 9.3.3.5. • Memo, Page 7, Table 4. The 8961, 9951, and 9971 IP phones were updated from version 9.2.1 to 9.4.1.
LEGEND: IP Internet Protocol NA Not Applicable			